

(19)



Europäisches Patentamt
European Patent Office
Office européen des brevets



(11)

EP 1 018 854 A1

(12)

EUROPEAN PATENT APPLICATION

(43) Date of publication:
12.07.2000 Bulletin 2000/28

(51) Int Cl.7: H04R 25/00, G10L 21/02,
H04R 3/00

(21) Application number: 99610002.0

(22) Date of filing: 05.01.1999

(84) Designated Contracting States:
AT BE CH CY DE DK ES FI FR GB GR IE IT LI LU
MC NL PT SE
Designated Extension States:
AL LT LV MK RO SI

(71) Applicant: OTICON A/S
2900 Hellerup (DK)

(72) Inventor: Hansen, Kim Vejlby
2900 Hellerup (DK)

(54) A method and a device for providing improved speech intelligibility

(57) The invention relates to a method for reduction of noise in an audio signal containing noise and a target signal, the method comprising, providing at least two input signals; processing the input signals by means of an independent component analysis, hereby determining

statistical dependencies of signal elements of the two input signals and determining whether statistical dependent signal elements form part of the target signal; outputting a part of the audio signal. The invention further relates to a device for use in reducing noise in an audio signal containing noise and a target signal.

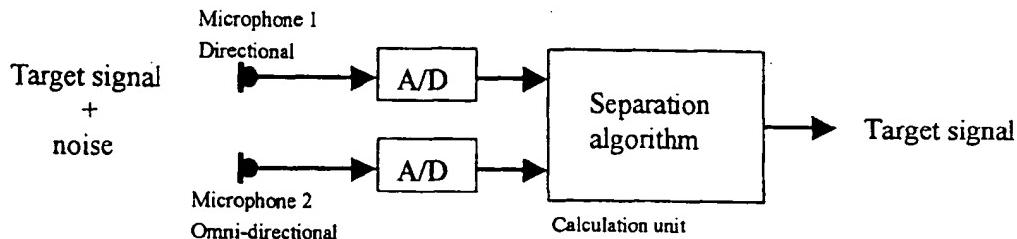


FIG. 2

to, Japan, Sept 4-6, 1996.

10 J.F. Cardoso, "Equivariant adaptive source separation", IEEE Transactions on Signal Processing 44(12), 1996.

11 S. Amari, A. Cichocki, H. H. Yang, "A new learning algorithm for blind signal separation", Advances in neural information Processing systems 8. MIT Press, 1996.

[0009] The objective of ICA is to recover the underlying independent source signals given only sensor observations that are linear mixtures of the original source signals. The only assumption of ICA is that the original source signals are statistically independent, otherwise the statistics of the source signals and the mixing of these into the sensor signals may be unknown. In contrast to correlation-based transformations such as Principal Component Analysis (PCA, I.T. Jolliffe, Principal Component Analysis, 1986, Springer Verlag), which decorrelates signals according to 2nd-order statistics, ICA also reduces higher-order statistical dependencies, in terms of maximising joint output entropy, in order to extract statistically independent signal components.

[0010] In the linear blind signal separation problem, N signals,

$$\mathbf{s}(t) = [s_1(t), \dots, s_N(t)]^T,$$

are mixed so that an array of N sensors picks up a set of signals

$$\mathbf{x}(t) = [x_1(t), \dots, x_N(t)]^T,$$

each of which has been mixed, delayed and filtered as follows

$$x_i(t) = \sum_{j=1}^N \sum_{k=0}^{M-1} a_{ijk} s_j(t - D_{ij} - k)$$

where D_{ij} are entries in a matrix of delays and a_{ijk} are the M -tap filter coefficients between the j th source and the i th sensor. The problem is to invert this scrambling without knowledge of it, thus recovering the original signals $\mathbf{s}(t)$ given only the $\mathbf{x}(t)$ signals. Finding this inverse scrambling is a challenging task since no informations are provided about the mixing nor the signals (hence the term blind separation). The type of architecture chosen for inverting the scrambling is important and can be made in numerous ways. An accurate architecture for inverting a M -tap filter is an infinite impulse response (IIR) filter with M coefficients. However, IIR filters are limited to have poles inside the unit circle, which imply that

a stable filter only exists for a minimum phase system. FIR filters may be used to approximate the inverse solution. Thus the inverse scrambling is performed according to

$$u_i(t) = \sum_{j=1}^N \sum_{k=0}^{M-1} w_{ijk} x_j(t - d_{ij} - k)$$

which has filters, w_{ijk} , and delays d_{ij} , which supposedly reproduce, at the output $\mathbf{u}(t)$, the original uncorrupted source signals, $\mathbf{s}(t)$, apart from a scaling factor for each signal and a permutation of signals.

[0011] Several algorithms have been proposed for the blind separation of linear mixtures. Bell and Sejnowski (3) proposed to learn the separating process by minimising the mutual information between components of $\mathbf{y}(t) = g(\mathbf{u}(t))$, where g is a non-linear function approximating the cumulative probability density function of the sources. They showed that for positively kurtotic signals (like speech) minimising the mutual information between components of $\mathbf{y}(t)$ is equal to maximising the entropy of $\mathbf{y}(t)$, which can be written as

$$H(\mathbf{y}) = -E[\ln(f_y(\mathbf{y}))],$$

where $f_y(\mathbf{y})$ denotes the probability density function of $\mathbf{y}(t)$. Denoting the determinant of the Jacobian of the whole unmixing process by $|J|$, $f_y(\mathbf{y})$ can be written as $f_y(\mathbf{x})/|J|$ (the Jacobian is a matrix with entries of $\partial y_j / \partial x_i$).

Maximising the entropy of the output leads to maximising

$$E[\ln(|J|)],$$

which in turn can be developed into a stochastic gradient ascent rule using instances of $\mathbf{x}(t)$ and $\mathbf{y}(t)$, instead of using the expectation. Thus

$$\Delta \mathbf{W} \propto (1 - 2\mathbf{y}(t))\mathbf{x}(t)^T + [\mathbf{W}^T]^{-1}$$

where $g(u) = 1/(1 + e^{-u})$ is used to approximate the cdf.

[0012] The algorithm can be made more efficient and independent of the conditioning of the mixing process (matrix) by using the so-called natural gradient instead of the absolute gradient, see Amari (11).

[0013] One particular proposed application of ICA is within electroencephalographic (EEG) recording of scalp potentials in humans and related brain activity

put signals according to a predetermined scheme.

[0027] Due to the fact that most hearing impaired have a hearing disorder which makes it even more difficult than for normal hearing persons to separate a target signal from the noise, which is often present in a speech situation, the invention is particularly relevant in connection with the technical field of hearing aids.

[0028] The invention therefor further relates to a hearing aid comprising: at least two microphones for audio signal input; signal processing means in connection with the microphones; an amplifier in connection with the signal processing means; a receiver in connection with the amplifier for outputting a signal from the amplifier; the signal processing means being adapted to process the signals by means of an independent component analysis method or a similar method based on the input from the at least two microphones, the processing comprising determining whether statistical dependent signal elements are present and removing at least part of the unwanted signal elements, thereby enhancing other parts of the audio signal.

[0029] The hearing aid according to the invention may further comprise the features set forth above, either separate or in combination.

[0030] Other fields of relevant use of the invention may be telecommunication or audio systems. In such systems the input and output may be connected to antennas or similar transmission and receiving means or may comprise microphones as input means as in the case of a hearing aid. Other elements of such systems may be standard elements, as these are not influenced by the signal processing according to the invention.

[0031] The invention will be described more detailed with reference to the accompanying drawings.

BRIEF DESCRIPTION OF THE DRAWINGS

[0032]

- Fig. 1 is a diagram showing the principles of the invention;
- Fig. 2 is a schematic diagram showing the principles of an implemented version of the invention;
- Fig. 3 is a schematic diagram showing the principles of an implemented version of the invention;
- Fig. 4 is a schematic diagram showing the implemented version of the invention as shown in fig. 2 further implemented in a hearing aid.

DESCRIPTION OF THE PREFERRED EMBODIMENT

[0033] The fundamental principle of the invention is schematically shown in fig. 1. The invention is basically a system, e.g. a hearing aid, with two or more sensors and a calculation unit. The calculation unit carries out the separation of the target and noise signals, by using the independence of the mixed signals according to ICA or a similar method comprising the basics of the ICA.

The sensors are arranged so that one is positioned to receive sound primarily from a target direction in front, whereas the others have arbitrary characteristics that do not specifically favour the target direction. Hereby, it is possible to use the technique of independent component analysis to separate the desired signal, which impinges from the target direction, from the disturbing noise signals, which impinge from any other directions.

[0034] To illustrate the invention, an example is given of a system implementing signal processing as described above. Fig. 2 schematically shows the signal processing system. The system comprises a directional and an omni-directional microphone, and a digital signal processing unit implementing the signal separation algorithm. Using the directional microphone gives the target direction from in front of the user, whereas the omnidirectional microphone gives a signal equally representing all signals around the head of the user.

[0035] A particularly important property of the independent component analysis is that it separates convolved and delayed source signals, where each independent source signal is defined as a signal which appears in the same way within each mixing process. Another important characteristic about the independent component analysis is that knowledge about the ratio of the source signals within the mixed signals can be used for classifying the separated signals. If for instance one source signal appears with a significantly better signal to noise ratio in one of the sensor signals, this information can be used to ensure that this source signal always will appear in a fixed output. Within the present invention these two characteristics combined with an appropriate placement of at least two sensors are exploited to eliminate signals not coming from in front of the user of the device.

[0036] From fig. 3 an embodiment appears, which comprises the features of the embodiment of fig. 2, but where the signal processor produces two output signals. By means of switching means one of the two output signals may be selected for further processing, e.g. amplification, or for output.

[0037] From fig. 4 an embodiment appears as a hearing aid according to the invention. The essential components of the hearing aid comprise two microphones, preferably a directional microphone and an omni-directional microphone, and an A/D converter connected to each of the microphones. The A/D converters are connected to a digital signal processor, which is adapted to perform the ICA method on the incoming signals. The signal from the signal processor is then lead to an amplifier and from this through a D/A converter to a receiver for performing the output of the processed signal. The devices of the figs. is in a usual manner powered by means of usual power sources, such as batteries.

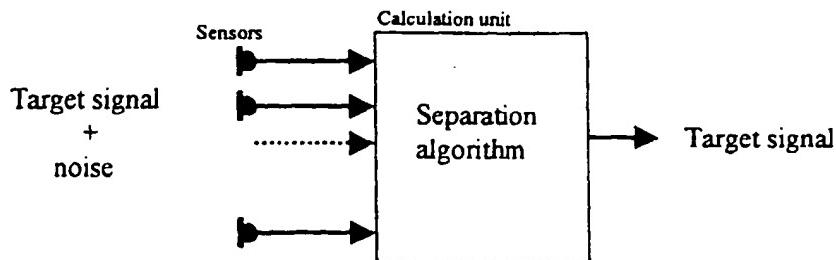


FIG. 1

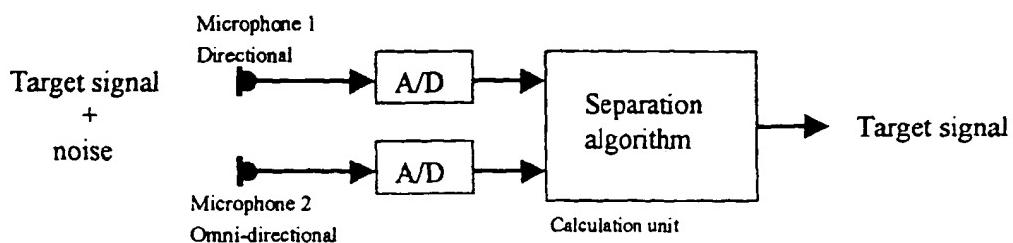


FIG. 2

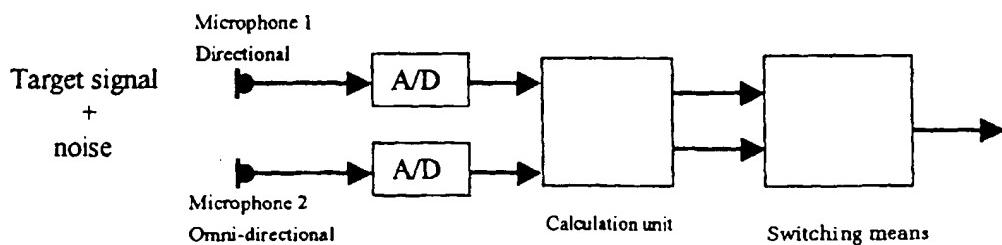


FIG. 3



European Patent
Office

EUROPEAN SEARCH REPORT

Application Number

EP 99 61 0002

DOCUMENTS CONSIDERED TO BE RELEVANT			CLASSIFICATION OF THE APPLICATION
Category	Citation of document with indication, where appropriate, of relevant passages	Relevant to claim	
X	WO 98 25214 A (ZIEHE ANDREAS ;GMD FORSCHUNGSZENTRUM INFORMAT (DE); MUELLER KLAUS) 11 June 1998 (1998-06-11) * page 2, line 4 - page 10, line 15 * * page 29, line 6 - line 10 *	1	HO4R25/00 G10L21/02 H04R3/00
Y	US 5 524 056 A (KILLION MEAD ET AL) 4 June 1996 (1996-06-04) * column 3, line 37 - line 49; figures * * column 8, line 16 - column 9, line 30 *	10	
A	DE 195 31 388 C (EHLERS FRANK ;SCHUSTER HEINZ GEORG PROF DR (DE)) 25 July 1996 (1996-07-25) * page 8, line 5 - line 11 *	1-3,6-9, 11	
X	US 5 208 786 A (FEDER MEIR ET AL) 4 May 1993 (1993-05-04) * column 1, line 60 - column 3, line 51; figures *	1,5	
A	EP 0 565 479 A (UNIV RAMOT) 13 October 1993 (1993-10-13) * page 4, line 58 - page 5, line 51 *	1,5	G10L H04R
A	TANIGUCHI T ET AL: "Blind signal separation for recognizing overlapped speech" J. ACOUST. SOC. JPN. (E) (JAPAN), JOURNAL OF THE ACOUSTICAL SOCIETY OF JAPAN (E), NOV. 1998, ACOUST. SOC. JAPAN, JAPAN, vol. 19, no. 6, pages 385-390, XP002108812 ISSN: 0388-2861 * abstract *	1,5	
		-/-	
The present search report has been drawn up for all claims			
Place of search	Date of completion of the search	Examiner	
THE HAGUE	12 July 1999	Gastaldi, G	
CATEGORY OF CITED DOCUMENTS			
X : particularly relevant if taken alone Y : particularly relevant if combined with another document of the same category A : technological background D : non-written disclosure P : intermediate document		T : theory or principle underlying the invention E : earlier patent document, but published on, or after the filing date D : document cited in the application L : document cited for other reasons & : member of the same patent family, corresponding document	

**ANNEX TO THE EUROPEAN SEARCH REPORT
ON EUROPEAN PATENT APPLICATION NO.**

EP 99 61 0002

This annex lists the patent family members relating to the patent documents cited in the above-mentioned European search report.
The members are as contained in the European Patent Office EDP file on
The European Patent Office is in no way liable for these particulars which are merely given for the purpose of information.

12-07-1999

Patent document cited in search report		Publication date		Patent family member(s)	Publication date
WO 9825214	A	11-06-1998	DE	19652336 A	04-06-1998
US 5524056	A	04-06-1996	EP	0664071 A	26-07-1995
			WO	9424834 A	27-10-1994
DE 19531388	C	25-07-1996		NONE	
US 5208786	A	04-05-1993	IL	102738 A	04-08-1996
			WO	9305503 A	18-03-1993
EP 0565479	A	13-10-1993	IL	101556 A	04-08-1996
			JP	6021838 A	28-01-1994
			US	5539832 A	23-07-1996
US 5706402	A	06-01-1998	AU	4596596 A	19-06-1996
			WO	9617309 A	06-06-1996